LIMP

Program for Loudspeaker Impedance Measurement

User Manual

Version 1.8.5

Ivo Mateljan

Artalabs
J. Rodina 4,
21215 Kaštel Lukšić, Croatia

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1 What is *LIMP*?

The *LIMP* is a program for the measurement of the loudspeaker impedance and estimation of loudspeaker physical and dynamical parameters (also called Thiele-Small parameters). It is also a general purpose program for measuring impedance in the range from 1 to 200 ohms.

Requirements to use the *LIMP* are:

- Operating systems: Windows XP / Vista / 7 / 8
- Processor class Pentium, clock frequency 600 MHz or higher, memory 256M for Windows 2000/XP or 2MB for Vista/Windows 7
- Full duplex soundcard with synchronous clock for AD and DA converters
- WDM or ASIO soundcard driver (ASIO is trademark and software of Steinberg Media Technologies GmbH).

The installation of program is through common setup program for the ARTA Software.

All Windows registry data for *LIMP* will be automatically saved at first program execution, and files with extension ".LIM" will be registered to be opened with a program *LIMP*.

Results of measurement can also be saved as ASCII formatted files in .ZMA format.

The *LIMP* does not dump graphs to the printer, instead of this all graphs can be copied to the Windows Clipboard and pasted to other Windows applications.

2 Impedance Measurement Theory

2.1 Basic Circuit for Impedance Measurement

The measurement of the loudspeaker impedance is based on the system shown in Fig. 2.1. The reference resistor *R* is connected between the signal generator and a loudspeaker impedance Z.

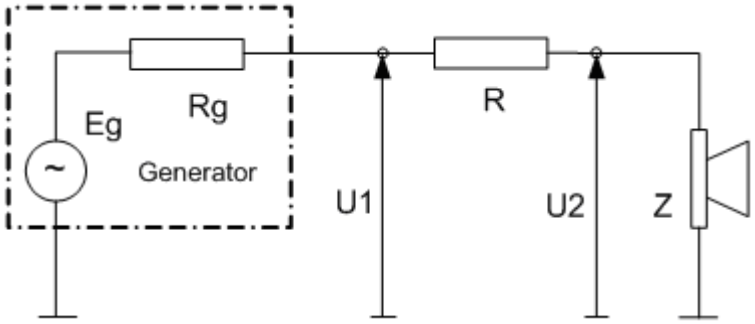


Figure 2.1 A circuit for loudspeaker impedance measurement

Impedance is defined in the frequency domain - $Z(f)$. If we measure voltages $U_1(f)$ and $U_2(f)$, at both ends of the reference resistor, we estimate the loudspeaker impedance as:

$$Z(f) = \frac{U_2(f)}{U_1(f) - U_2(f)} R \tag{1}$$

2.2 Stepped Sine and Periodic Pink Noise Generator

*T*he *LIMP* has two modes of impedance measurement:

1) Stepped sine mode
2) FFT mode with pink periodic noise excitation (Pink PN)

In the stepped sine mode *LIMP* generates bursts of pure sinusoidal signal, frequency by frequency with 1/6, 1/12, 1/24 or 1/48 octave increment, and measures response to sinusoidal signal by filtering out noise and distortion components in sinusoidal response.

FFT mode is faster method for impedance measurement, but with much lower measurement S/N ratio than stepped sine mode. In this mode the *LIMP* simultaneously measures values of the impedance in the whole audio range of frequencies by using the signal generator with a wideband periodic pink noise excitation (pink PN).
The wideband pink PN excitation is realized as periodic random phase multisine signals. It is a zero dc, periodic signal that contains *M* sine components, each with a random phase:

$$g(t) = \sum_{k=1}^{M} A_k \cos(2\pi k f_0 t + \varphi_k), \quad \varphi_k = random \in [0, 2\pi] \tag{2}$$

This pink multisine has $A_k = \sqrt{2}A_{2k}$ and its spectral magnitudes roll-off 3dB/oct (after specified cut-off frequency). In the *LIMP*, a variable low frequency cut-off frequency can be changed in the '**Signal Generator Setup**' dialog box. The pink noise is usually used with a cut-off frequency set close to the frequency where loudspeaker has an impedance maximum (20-100Hz). The pink periodic noise has the statistical distribution close to the normal distribution, and in *LIMP* it is generated with a crest factor lower than 12 dB.

2.3 Simple Measurement Procedure

In **Stepped sine mode**, a simple measurement procedure is as follows:

1. The *LIMP* generates burst of sine signal with frequency *f*. We assume that this signal drives the generator shown in Fig. 2.1.
2. After an arbitrary *transient time*, that is necessary to reach the steady state, voltages at both ends of the reference resistor are measured as discrete time series u_1 and u_2 of length *N*.
3. The magnitude and phase of signal u_1 and u_2 at frequency *f* are estimated by finding fundamental sinusoidal components U_1 and U_2 at generator frequency f. It is done by directly solving Fourier integral in *integration time* T. These fundamental harmonics are used in the equation (1) to calculate the impedance *Z*.
4. After some arbitrary time, we call it *intra burst pause*, that is necessary for measured system to release reactive energy, a frequency is incremented for 1/24 octave (or 1/48 octave) and the process is being repeated from step 1) until the predefined stop frequency is reached.

In **FFT mode**, a simple measurement procedure is as follows:

1. The *LIMP* generates the Pink PN as a discrete periodic sequence with a period equal to *N*. We assume that this sequence drives the generator shown in Fig. 2.1.
2. After one preaveraging cycle, which is necessary to reach the steady state, voltages at both ends of the reference resistor are measured as discrete time series u_1 and u_2 of length *N*.
3. The DFT is applied to time series u_1 and u_2 to get spectral components $U_1(f)$ and $U_2(f)$. They are used in the equation (1) to calculate the impedance $Z(f)$.

Important note:
The signal generation in the stepped sine mode gives sinusoidal components with at least 30dB higher level than it is possible in FFT mode. That is why, in FFT mode the measurement results can be greatly affected by the noise that can be generated by the measured loudspeaker.

2.4 Measurement in a Noisy Environment

The main source of the measurement noise is a loudspeaker that acts as a microphone for the environmental noise and vibrations. Fig. 2.2 shows modified circuit for the loudspeaker impedance measurement, with the noise generator E_n included.

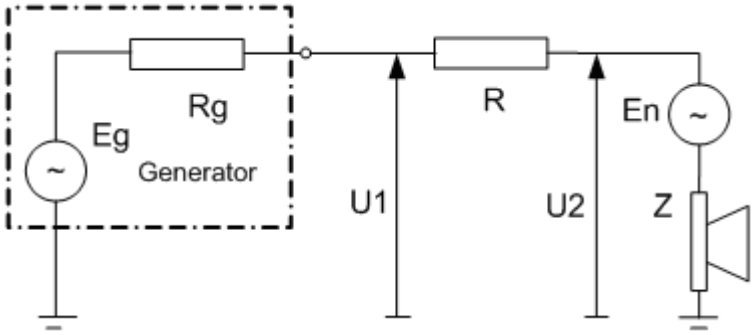


Figure 2.2 A circuit for the loudspeaker impedance measurement, with the noise generator E_n

If we apply equations for impedance measurement (1) to this circuit, we get the estimated impedance value:

$$Z_{estimated} = R\frac{U_2}{U_1 - U_2} = Z + \frac{R + Z + R_g}{1 + E_g / E_n} \tag{3}$$

What this equation shows is that estimated impedance differs from the true impedance Z by the term that is dependant on the S/N ratio (E_g/E_n) and values of resistors R, R_g and the impedance Z. We can conclude:

1. The signal generator must supply a high voltage, to assure high S/N. In practice, we need a generator with at least 1V of peak output voltage. This can easily be achieved with stepped sine excitation.
2. When we use FFT method the measurement results are highly affected with noise. The loudspeaker acts as a microphone with a highest sensitivity in the region of the membrane resonance. It means the highest level of the noise is at low frequencies, so we must generate the signal with highest level at low frequencies, i.e. the pink noise.
3. Values of resistors R and R_g must be small, an optimum being a value close to the magnitude of the measured impedance. Practically, we can use R = 10-27 ohm to get a very good impedance estimation, but then we need a power amplifier to supply large current. If we use the soundcard headphone output as a signal generator, then we can use R = 47-100 ohms. If we use the soundcard line output as a signal generator, due to the limited current capability, we must use R>600 ohms. In that case we can't get a good estimation with FFT method, but we still can with stepped sine method.

2.4.1 Lowering Measurement Noise in Stepped Sine Mode

In stepped sine mode *LIMP* uses "heterodyned" principle to filter all spectral components that are out of the passband which is centered at the measured frequency. The bandwidth of the filter is equal to 1/T where T is *integration time* of the Fourier integral. For example, if we use integration time 200ms, then the width of the passband of the "heterodyne" filter is 5Hz.

2.4.2 Lowering Measurement Noise in FFT Mode with Periodic Noise Excitation

The noise can be partly lowered by using the averaging technique in the estimation of U_1 and U_2. *The* LIMP averages the auto-spectrum of U1 and the cross-spectrum of U_1 and U_2, giving the following estimated impedance:

$$Z_{esitimated} = \frac{R}{\frac{\langle U_1 U_1^* \rangle}{\langle U_2 U_1^* \rangle} - 1} \tag{4}$$

Note: brackets <> denotes averaged values and star denotes the conjugate complex value.

Two types of averaging can be applied in the *LIMP*. In a "time domain synchronous averaging" the LIMP generates multiple periods of one noise sequence (Fig. 2.3). In a "frequency domain asynchronous averaging" the *LIMP* generates a different noise sequence for every acquired FFT block (Fig. 2.4). The Frequency Domain Asynchronous Averaging can give slightly better results than the synchronous averaging in systems with nonlinear distortion, but it needs longer time for measurements.

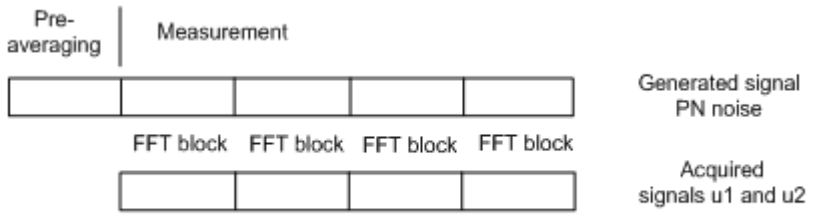


Figure 2.3 Signal generation and acquisition during the synchronous time domain averaging process

Figure 2.4 Signal generation and acquisition during the asynchronous frequency domain averaging

3. Hardware Setup

The simplest measurement configuration is shown in Fig. 3.1. The soundcard phone-out (headphone output) is used as a signal generator output. The soundcard left line-input is used for recording the voltage U_2 and the soundcard right line-input is used for recording the voltage U_1.
If the soundcard has no headphone output , then we have to connect external power amplifier to the soundcard line-out output, as shown in Fig. 3.2.

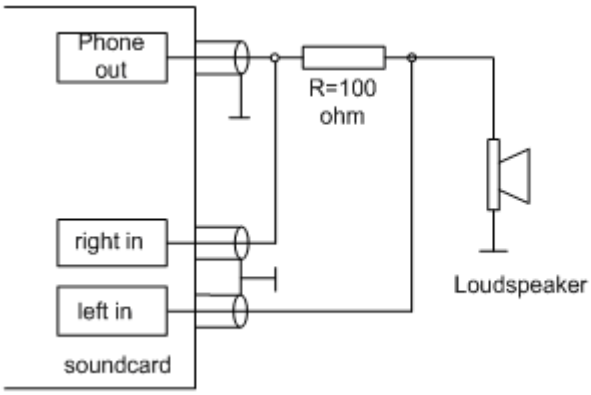


Figure 3.1 Measurement setup for loudspeaker impedance measurements, using the headphone output

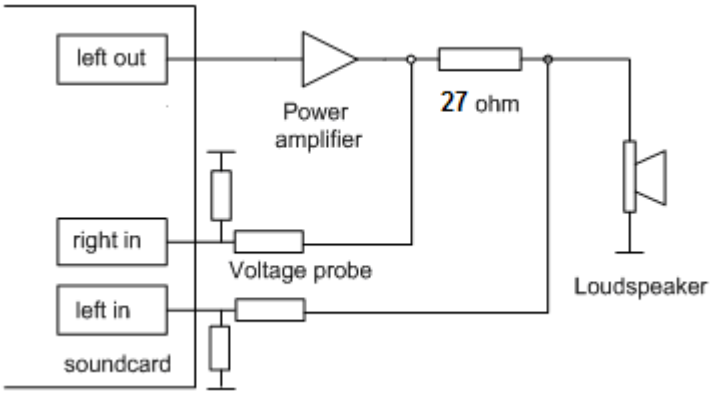


Figure 3.2 General measurement setup for loudspeaker impedance measurements

To protect the soundcard input from large voltage that is generated at a power amplifier output, it is recommended to use the voltage probe circuit with Zener diodes, as shown on Fig. 1.3. Values of resistors R1 and R2 have to be chosen for arbitrary attenuation (i.e. R1=8200 and R2=910 ohms gives probe with -20.7dB (0.0923) attenuation if the soundcard has usual input impedance - 10kΩ).

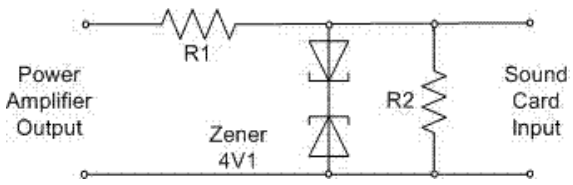


Figure 3.3 Voltage probe with the soundcard input channel overload protection

4 Working with *LIMP*

When you start the *LIMP* you get the program window shown in Fig. 4.1. There are: menu bar, toolbar and a dialog bar at the top of the window and a status bar on the bottom of the window. The central part of the window will show the magnitude and the phase plot.

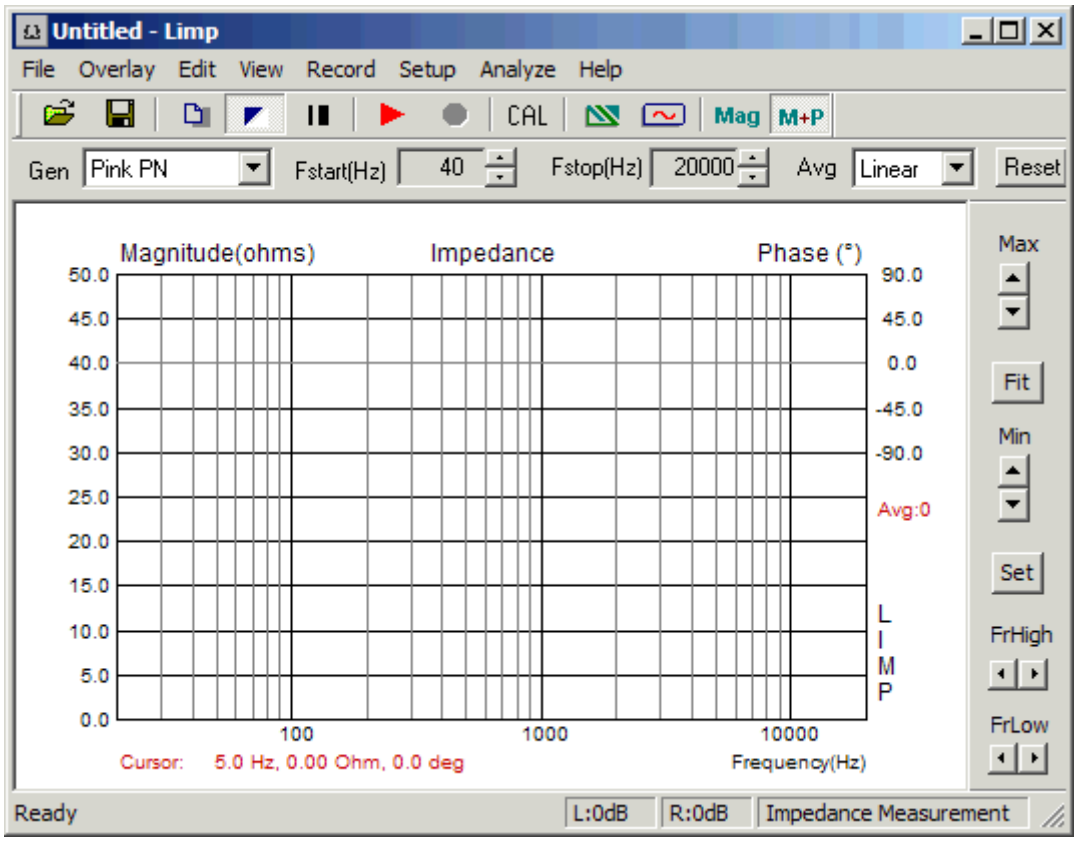


Figure 4.1 Main program window

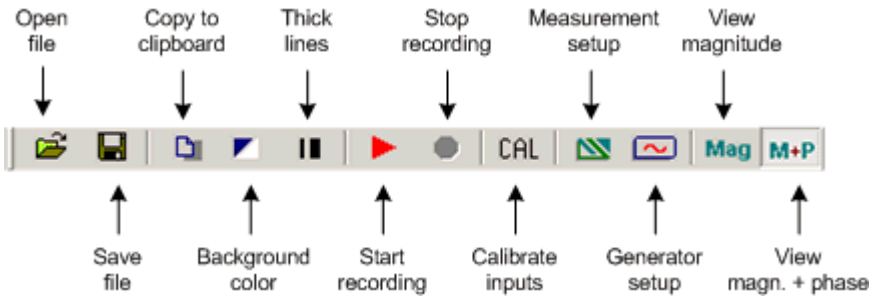
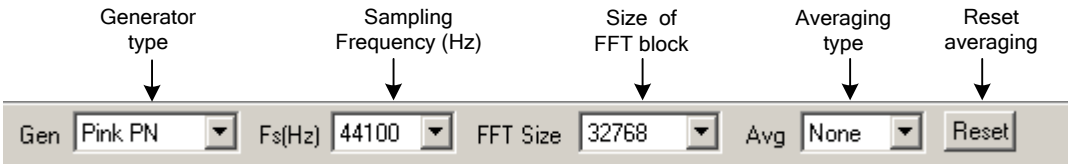


Figure 4.2 Toolbar icons

Figure 4.3 Status bar shows the peak level (ref. full scale) of left and right line inputs



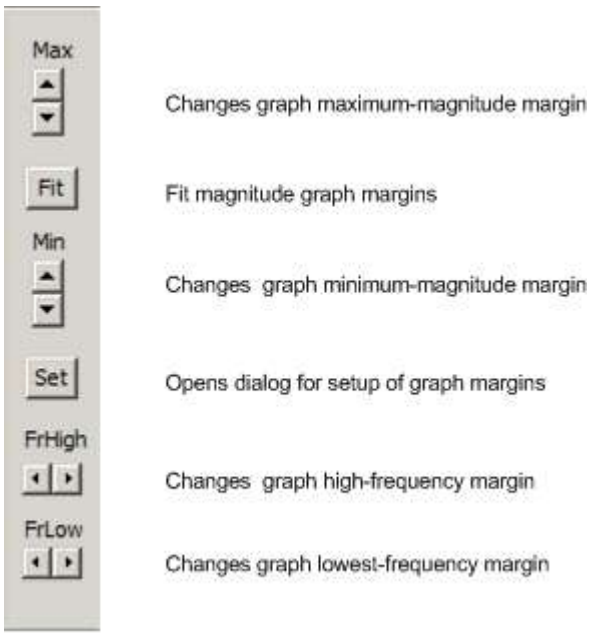


Figure 4.4 Top and right dialog bars

Images of graphs and windows can be copied to Windows clipboard or saved to the file in a three image formats: **png, bmp** and **jpg**. It is recommended to use **png** format.

Obtaining copy of the full window picture is simple. User needs to simultaneously press keys **Ctrl+P**. After that command the window picture will be saved in the System Clipboard, from were the user can paste it in other opened Windows applications (MS Word, MS Paint). Keys **Ctrl+Alt+P** activate command to save that image in the file.

To copy or save the graph picture, that is shown inside the window, user needs to simultaneously press keys **Ctrl+C** or activate the menu command '**Edit->Copy**', or press appropriate '**Copy**' button. In the main window toolbar, the '**Copy**' button is shown as toolbar icon .

The **Copy** command opens the dialog box '**Copy/Save Image with Extended Information**', shown in Figure 4.5. Here user has to setup following options:

1) By using the combo box above 'OK' button, user chooses one of three modes of saving the image: **Copy to Clipboard**, **Save to File** and **Save to File + Copy to Clipboard**.

2) In the **Edit box** user optionally enters the text that will be appended at the bottom of the graph.

3) Check box ' **Add filename and date**' enables adding text to the graph that shows file name, date and time.

4) Check box '**Save text**' enables saving entered text for the next copy operation.

5) **Bitmap size** is chosen by selecting one of following combo box items:
   - Current screen size - variable width and height option
   - Smallest (400 pts) - fixed graph width 400 points
   - Small (512 pts) - fixed graph width 512 points
   - Medium (600 pts) - fixed graph width 600 points
   - Large (800 pts) - fixed graph width 800 points
   - Largest (1024 pts) - fixed graph width 1024 points

   The options with fixed width give graphs with the aspect ratio 3:2.

   The button '**OK**' copies the graph to the system clipboard or opens dialog to enter name of file in which picture will be saved.
   The button '**Cancel**' cancels the copy operation.

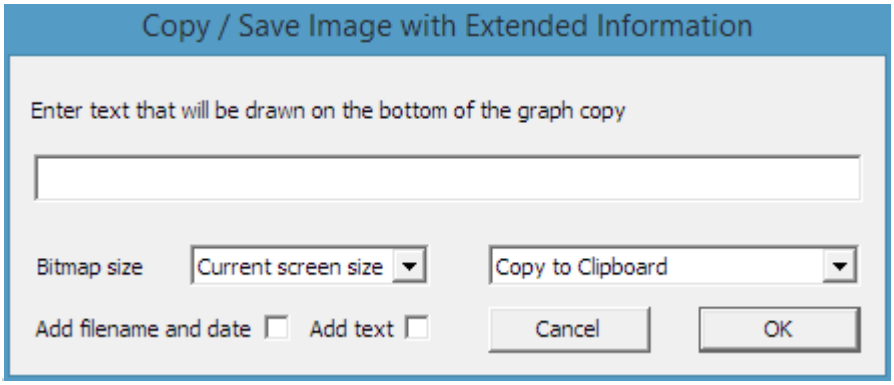

Copy / Save Image with Extended Information
Enter text that will be drawn on the bottom of the graph copy
Bitmap size
Current screen size
Copy to Clipboard
Add filename and date
Add text
Cancel
OK

Figure 4.5 Dialog box '**Copy / Save with Extended Information**',

4.1 LIMP Menus

Here is a brief explanation of *LIMP* menus:

File

New - initializes a new file content
Open... - opens an existing impedance file (*.LIM)
Save - saves the file with a current name
Save As... - saves the file under a new name
Info. - shows information about current file
Export ASCII - exports the file content in a textual - ASCII file
Commented .TXT file - exports textual file with comment and rows containing frequency, impedance magnitude and phase
Plain .ZMA file - exports textual file with rows containing values of frequency, impedance magnitude and phase
CSV (Excel) file - exports CSV file containing frequency, impedance magnitude and phase

Recent File - opens a recently opened file
Exit - exits the program

Overlay

Set as overlay- sets the current impedance curve as an overlay
Delete - deletes overlay
Load - loads an overlay from an impedance file
Set as target curve - sets current curve as target curve
Load target curve - loads an target curve from an impedance .lim or .zma ofile
Delete target curves - deletes all target curves

Edit

Copy - copies the graph bitmap and user defined text to the clipboard or saves that image to the file
Setup colors and grid style - sets graph colors and grid style
B/W Color - changes the background color to black or white
Use thick pen - uses thick pen for plotting curves
Subtract overlay - subtracts overlay impedance from current impedance
Add overlay - add overlay impedance to current impedance

View

Toolbar - shows or hides the Toolbar
Status Bar - shows or hides the Status bar
Fit graph range - changes graph top margin to show full impedance curve
Magnitude - shows only impedance magnitude curve
Magnitude+phase - shows impedance magnitude and phase curves

Record

Start - starts recording (measurement)
Stop - stops recording (measurement)
Calibrate - opens the dialog box for calibration of line input channels

Setup

Audio devices - sets current input and output devices
Generator - sets generator parameters
Measurement - sets measurement parameters
Cable compensation – sets resistance and inductance for cable impedance compensation
Graph - sets graph margins
CSV format - opens dialog box for setup of decimal separator character in CSV files

Analyze

Loudspeaker parameters - Added mass method - shows the dialog box for the estimation of loudspeaker parameters using Added mass method
Loudspeaker parameters - Closed box method - shows the dialog box for the estimation of loudspeaker parameters using Closed box method
RLC impedance values at cursor position - estimate values of impedance serial RLC elements

Help

About - gets information about the *LIMP*
Registration - shows the user license registration
User Manual - shows the help file

Useful shorcut keys are:

Up and **Down** keys - change the top graph margin
Left and **Right** keys - move the cursor left and right
Ctrl+S key - saves the file
Ctrl+N key - makes a new file
Ctrl+O key - opens the file
Ctrl+C key - copies the graph bitmap and user defined text to the clipboard or saves that image to the file
Ctrl+P key - copies a whole window bitmap to the clipboard
Ctrl+Alt+P key - saves a whole window bitmap to the file
Ctrl+B key - changes background color
F3 key - make target curve by calculating loudspeaker impedance using current TSP parameters

4.2 Soundcard Setup

Activate the menu **Setup->AudioDevices**. You will get the '**Soundcard Setup**' dialog box shown in Fig. 4.6. In this dialog box you choose which soundcard will be used as an input or output device. Generally, choose the same card as an input and output device.

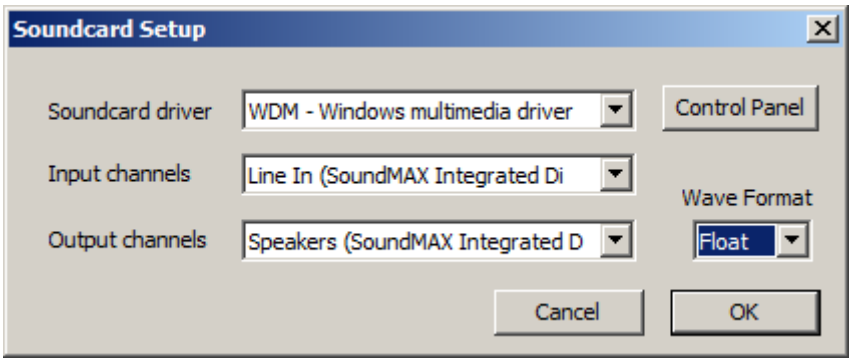


Figure 4.6 Soundcard setup

The '**Audio Device Setup**' dialog box has following controls:

Soundcard driver - chooses the type of soundcard driver (WDM – windows multimedia driver or one of installed ASIO drivers).
Input channels - chooses the soundcard input stereo channels. ASIO driver can have large number of channels.
Output Device - chooses the soundcard output stereo channels.
Generally, user chooses input and output channels of the same soundcard (mandatory in ASIO driver mode).
Control panel button – if WDM driver is chosen, it opens sound mixer on Windows 2000/XP or Sound control panel in Vista/Win7. If ASIO driver is chosen, it opens ASIO control panel.
Wave format – on Windows 2000/XP chooses Windows wave format: 16 bit, 24 bit, 32 bit or Float. Float means IEEE floating point single precision 32-bit format. It is recommended to use 24-bit or 32-bit modes when using a high quality soundcard (many soundcards are declared as 24-bit, but their real bit-resolution is less than 16-bits). For Windows Vista / Windows 7 it is recommended to choose resolution type Float. This control has no effect in the ASIO mode, where a bit resolution has to be setup in the ASIO control panel.

Important notice: Please mute the line and microphone channels at the output mixer of the soundcard; otherwise you might have positive feedback during measurements. If you use a professional audio soundcard, switch off the direct or zero-latency monitoring of the line inputs.

4.2.1 WDM Audio Driver Setup for Windows 2000 / XP

After selection of the soundcard user has to disable (mute) line-in and microphone inputs in output mixer. Also user has to select which input will be used for recording: Line In or Microphone (Mic). For a standard PC soundcards the procedure is as follows:

1. In '**Audio device setup**' dialog click the button '**Control panel'** to open the Windows '**Master Volume**' dialog box, which is shown on Fig. 4.8.
2. Click on menu '**Options->Property**' and select soundcard channel that will be used for output (playback), as shown in Fig. 4.7.
3. Mute Line In and Mic channels in dialog '**Master Volume**' (Fig. 4.8)
4. Set Master Volume and Wave Out volume to maximum.
5. Click on menu '**Option->Property**' and select soundcard channel that will be used for input and enable Line In and Mic channels in recording mixer.
6. Choose Line In Input.
7. Set volume control of Line In to some lower position. Later it will be set more precisely.

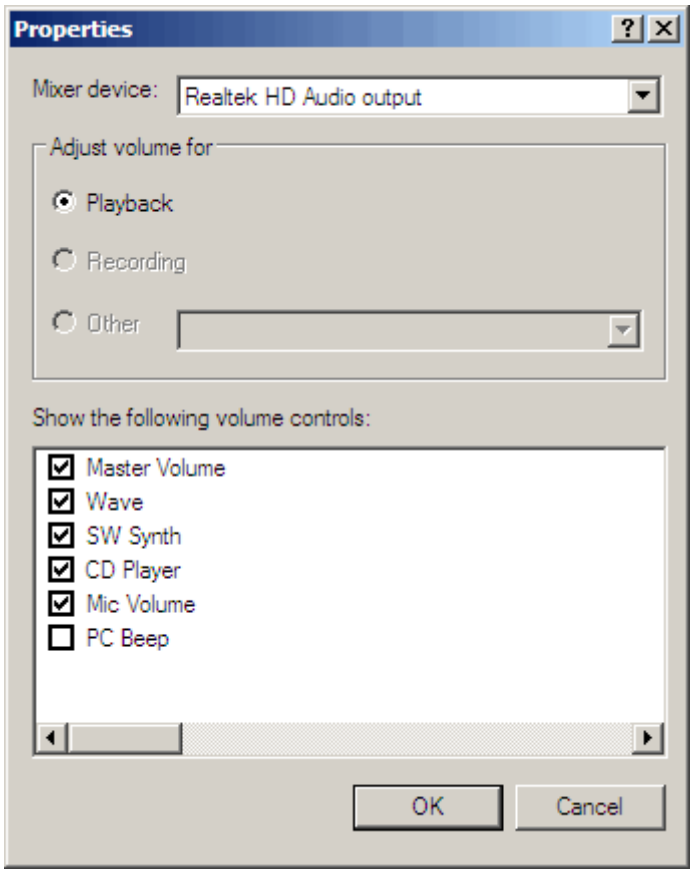


Figure 4.7 Dialog for choosing soundcard and input/output channels

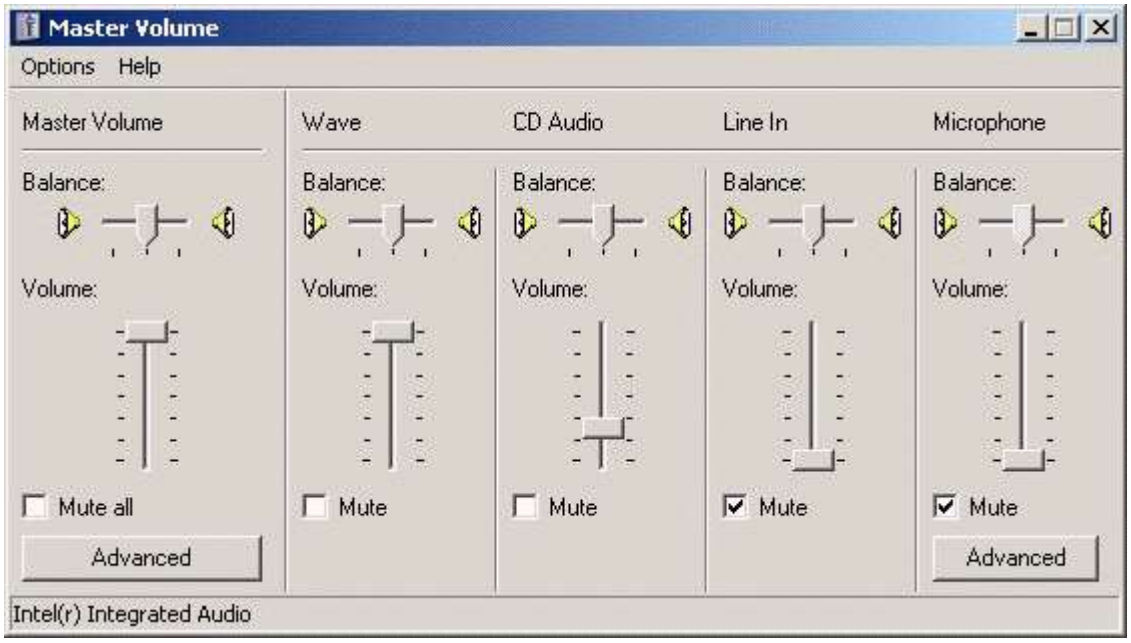


Figure 4.8 Typical setup of a soundcard output mixer in Windows XP

Note: Most professional audio soundcards have their own program for adjustment of input and output channel, or have hardware control of input monitoring, and input and output volume controls.

4.2.2 WDM Audio Driver Setup for Windows Vista / 7 / 8

Microsoft has changed their approach in control of sound devices in Vista / Win7. Now operating system (also, sometimes in conjunction with control programs of professional soundcards) is responsible for setting soundcard native sampling rate and bit resolution. Operating system changes native resolution to floating point format for high quality mixing and eventually for the sample rate conversion.
For LIMP, this means that it is strongly recommended to use '**Float**' resolution and sets the sampling rate to the native format. Access to these values is in '**Windows sound control panel**', which user gets by clicking on the button '**Control Panel**' in '**Audio Device Setup'** dialog**.**

Fig. 4.9 shows Vista/7 control panel, that has four property pages.

As first step, user has to adjust '**Playback page'** and later repeat the same procedure for '**Recording page'**. Adjustment steps are:

1) Click on channel info to choose a playback channel. It is not recommended to use the measurement channel as a default audio channel.
2) Click on button '**Properties**' to opens channel '**Sound properties**' dialog.
3) Click on the tab '**Levels**' to open the output mixer (as in Fig. 4.10). Then mute Line In and Mic channels, if exist.
4) Click on the tab '**Advanced**' to set the channel resolution and a sample rate (as in Fig. 4.11)
5) Repeat previous procedure 1) to 4) for recording channel, and choose the same sampling rate as in the playback channel.

Figure 4.9 Vista Sound Control panel

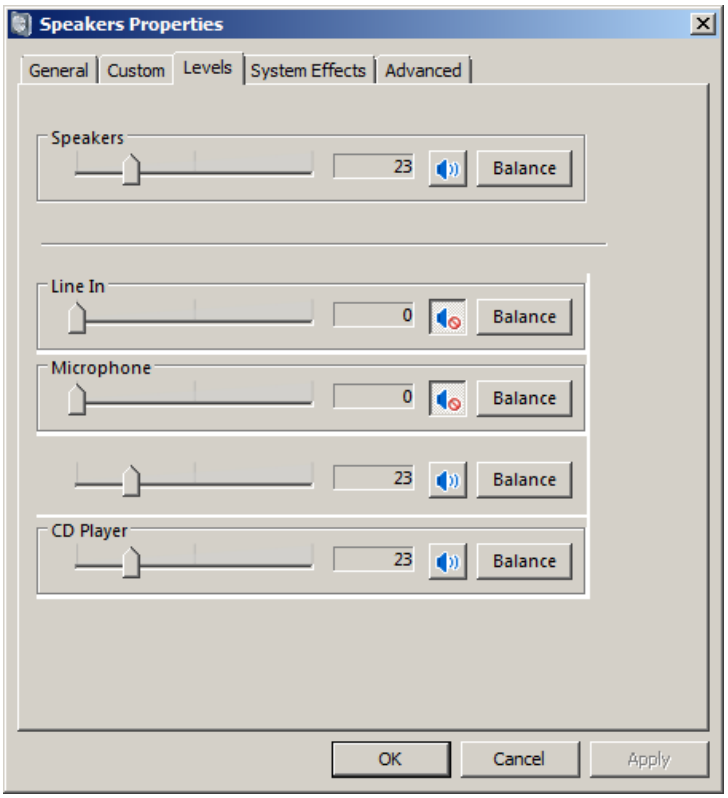


Figure 4.10 Playback channel properties – Output levels

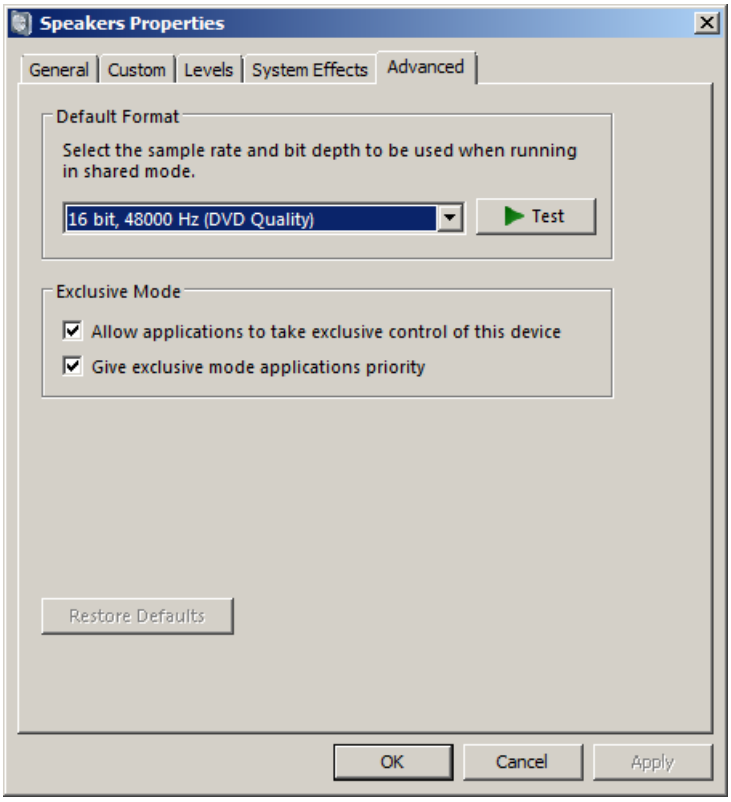


Figure 4.11 Setting the native bit resolution and sampling rate in Vista

Note: There are a lot of drivers that do not work stable under Windows 7. In that case please use ASIO driver if it is available for your soundcard.

4.2.3 ASIO driver setup

ASIO drivers are decoupled from the operating system control. They have their own control panel to adjust native resolution and memory buffer size. The buffer is used for transfering sampled data from the driver to the user program. User opens the ASIO control panel by clicking button '**Control Panel**' in the '**Audio Device Setup**' dialog. Fig. 4.12 shows an example of ASIO control panel.

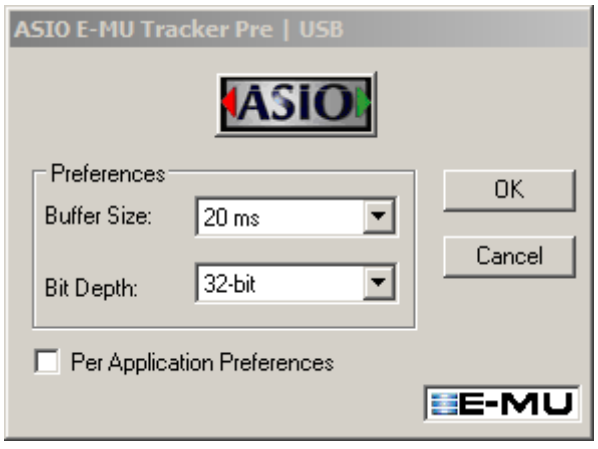


Figure 4.12 E-MU Tracker Pre ASIO Control panel for setting bit-resolution and buffer size

In music applications user usually sets buffer size as small as possible with stable operation. That gives the lowest input/output latency (system introduced delay).

In LIMP, the latency is not problem, as it is encountered in software, but it is not recommended to use buffer with size larger than 2048 samples, or smaller than 256 samples. Some ASIO control panels express the buffer size in samples, while other express the buffer size in time [ms]. In that case we can calculate the size in samples using the following expression:

buffer_size [samples] = buffer_size[ms] *samplerate[kHz] / number_of_channels.

Some ASIO drivers allow setup of buffer size (in samples) that is a power of number 2 (256, 512, 1024, …). In that case Limp adjusts buffer size automatically.

LIMP always works with two input channels, and two output channels, treating them as a stereo left and right channels. As ASIO support multichannel devices, user has to choose in a dialog box 'Soundcard **Setup**'' which pair of channels will be used (1/2, 3/4, …).

4.3 Generator Setup

Two types of excitation signals are possible in the *LIMP*:

- Sine
- Pink periodic noise (Pink PN).

Choose the signal type in the top dialog bar or in the **'Generator Setup'** dialog box shown in Fig. 4.13. You get it by clicking the menu **Setup -> Generator** or by clicking the toolbar icon .

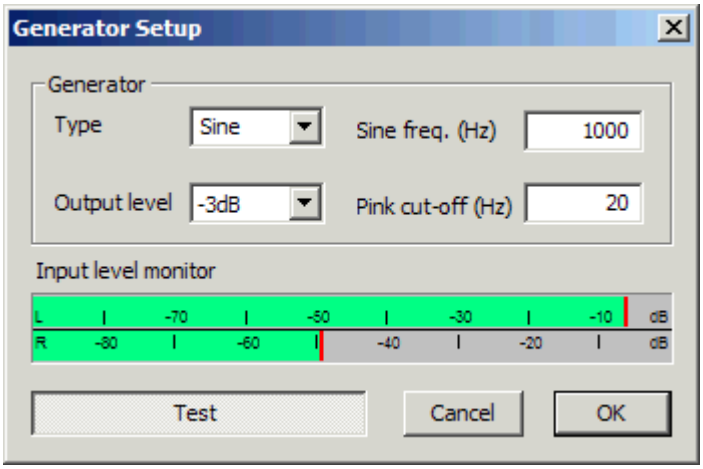


Figure 4.13 Signal generator setup

The dialog **Generator Setup** has following controls:

Type - chooses the excitation signal: **Sine** or **Pink PN**.
Sine freq. (Hz) - enter test frequency of the sine generator.
Pink cut-off (Hz) - enters the low-frequency cut-off of the pink noise generator.
Output level (dB) - chooses the output level (0-15dB).
Test - starts/stops generator with current settings.
Input level monitor - the input volume peak meter. Meter bars are shown in following colors:
green (for levels below -3dB),
yellow (for levels between -3dB and 0dB) and
red (for input overloaded).

Recommendation:

- For most reliable results use sine generator, but don't push loudspeaker into large displacement (the largest displacement is at frequencies below the loudspeaker resonance frequency).
- For measurements of bass or mid-bass loudspeakers set the Pink cut-off frequency close to the loudspeaker resonance frequency (20-100 Hz).
- Press the button **Test** to monitor input/output levels. If peak meter bars are in red color input channels are overloaded, then lower the output volume until bars go to yellow or green color.

4.4 Measurement Setup

For setup of measurement parameters use the **'Measurement setup'** dialog box, shown in Fig. 4.14. You get it by clicking the menu **Setup->Measurement** or by clicking the toolbar icon .

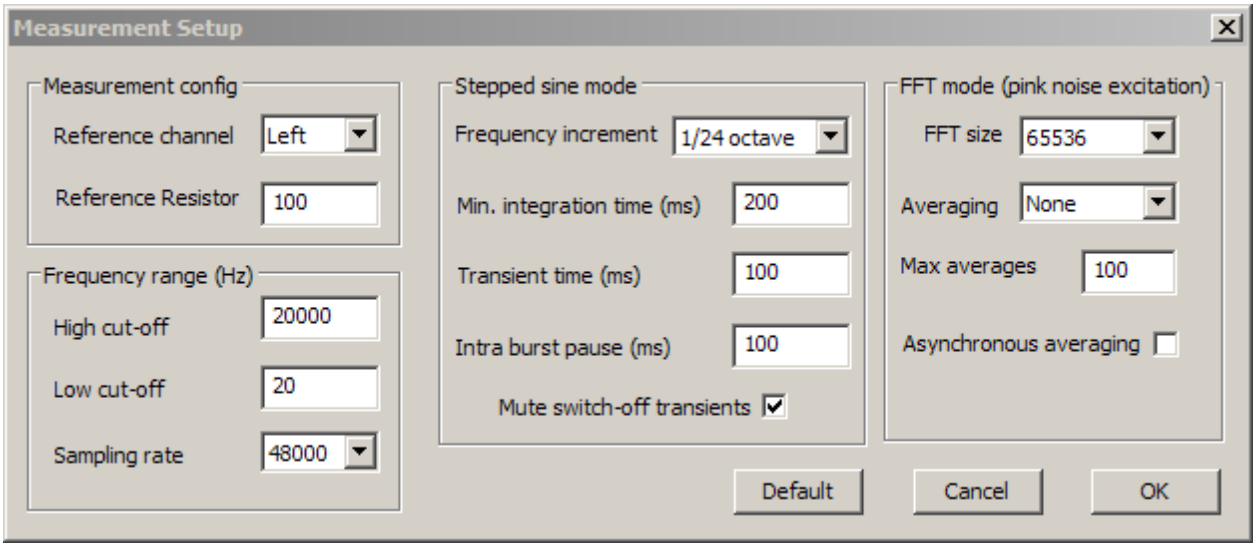


Figure 4.14 Measurement setup

The dialog ' **Measurement setup** ' has following controls:

In section **Measurement config:**

Reference channel - sets the reference channel (U_1) to: Left or Right.
Reference resistor - enters the value for the reference resistor.

In section **Frequency range:**

High cut-off - enters measurement upper frequency margin.
Low cut-off - enters measurement lower frequency margin.
Sampling rate (Hz) - chooses the sampling rate (from 8000 to 96000Hz).

In section **Stepped sine mode:**

Frequency increment - sets to 1/24 or 1/48 octave
Min. integration time (ms) - enters minimum integration time (higher values give higher noise reduction).
Transient time (ms) - enters transient time to allow the system to get to the steady state (before starting the integration).
Intra burst pause (ms) - enters time needed for system to release the energy from reactive components.

Mute switch-off transients – check box to remove annoying sound due to sine burst abrupt switch-off

In section **FFT mode (pink noise excitation):**

FFT size - chooses the length of the FFT block (32768 or 65536).
Type - chooses: None, Linear or Exponential averaging.
Max. Averages - enters the maximum number of averaging for the 'Linear' mode.
Asynchronous averaging - check box to use the asynchronous averaging.

4.5 Measurement Procedures

After you have done the audio device setup, the generator setup and the measurement setup you are ready for measurements.

a) Procedure in FFT mode

Connect loudspeaker in test fixture (Fig. 3.1 or 3.2) and click the menu **Record->Start** or click the toolbar icon . Measurements will be periodically repeated and results shown as an impedance plot. You will get the graph like one shown in Fig. 4.15. You can stop the measurements by clicking the menu **Record->Stop** or by clicking the toolbar icon .

You can copy the graph bitmap by clicking the menu **Edit->Copy** or by clicking the toolbar icon .

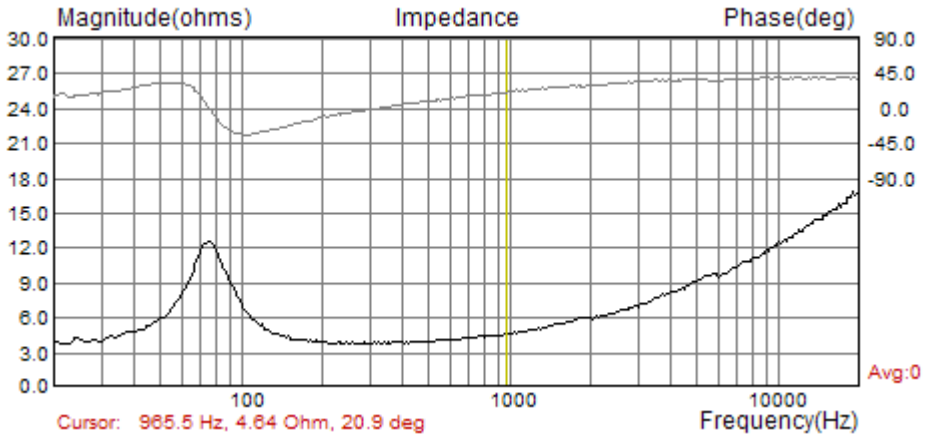


Figure 4.15 Impedance measurement without averaging – a noisy measurement

If you set averaging to **Linear**, measurements will be repeated and averaged until the number of averaging reach the predefined value for **Max averages** in the '**Measurement setup**' dialog box. You can stop the averaging at any time by clicking the menu **Record->Stop** or by clicking the toolbar icon . You will get the graph like one shown in Fig. 4.16.

You can also choose the exponential averaging. It differs from linear averaging in a way that it gives more weight to results from last five measurements.

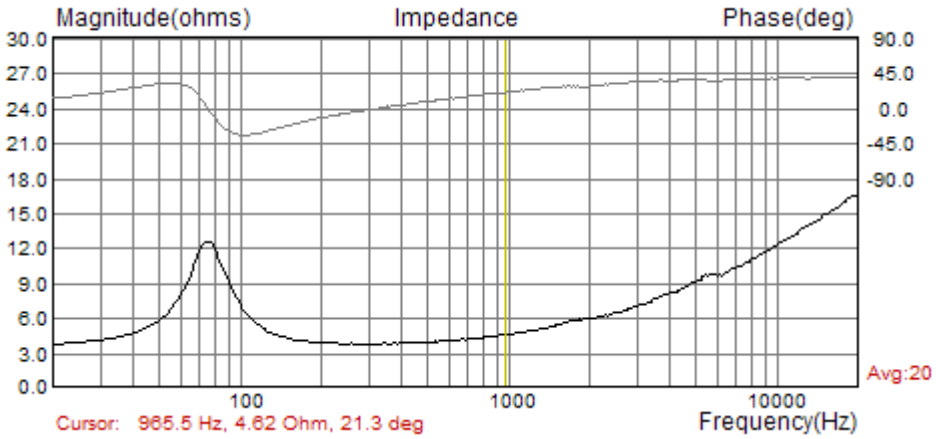


Figure 4.16 Impedance measurements with averaging

a) Procedure in Stepped sine mode

Measurement procedures for stepped sine mode are almost the same as in FFT mode. The only difference is that in FFT mode user will see the impedance plot for full measured frequency range almost instantly, while in stepped sine mode procedure will be repeated for many frequencies. Cursor will show current progress in measurements, and that process will be very slow.

4.6 Graph Setup and Browsing

The menu command **Setup->Graph Setup** (or by *right-clicking mouse in the plot area*), opens the **'Graph Setup'** dialog box, shown in Fig. 4.17. Use this dialog box to adjust the impedance magnitude range shown and the frequency range shown.

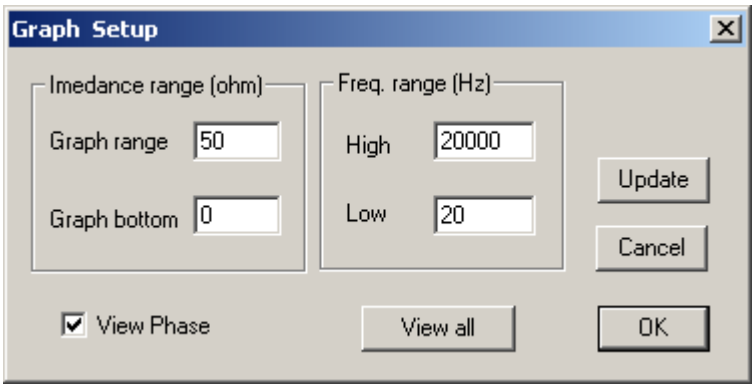


Figure 4.17 Graph setup

The '**Graph Setup**' dialog box has following controls:

Impedance range (ohm) section:

Graph range – enters the impedance magnitude range.

Graph bottom – enters the impedance magnitude for graph bottom margin.

Freq. range (Hz) section:

High – enters the highest frequency shown (in Hz).

Low – enters the lowest frequency shown (in Hz).

View All – enables the view of all DFTspectrum components that are used in impedance estimation.
View Phase – enables a phase plot.
Update – updates the graph with a new setup.

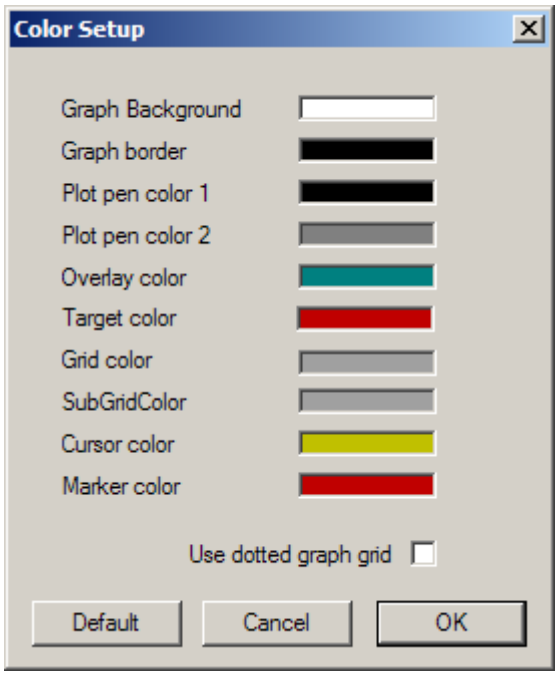


Figure 4.18 Dialog boxes for a graph colors setup

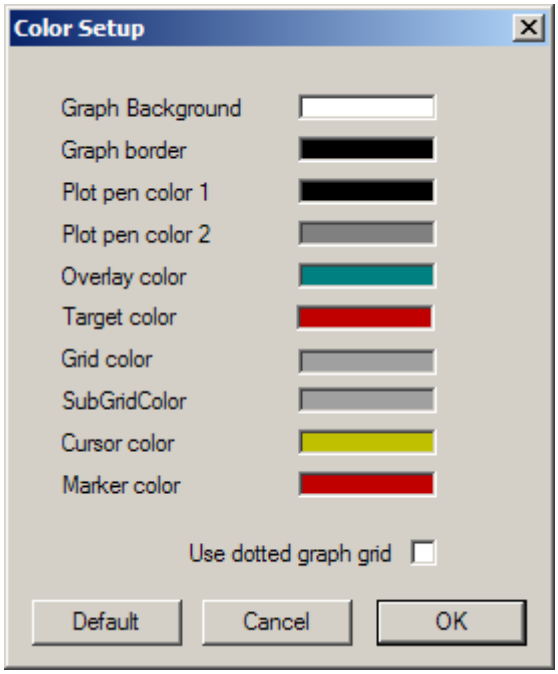


Figure 4.19 Standard Windows dialog for color setup

Graph colors can be changed in two ways. The first one is to change the background color from "Black" to "White" by clicking the menu command **View->B/W color** or by clicking the toolbar icon .
The second way to change graph colors is a "user mode". User sets an arbitrary color for every graph element using the '**Color Setup**' dialog box, shown in Fig. 4.18. This dialog can be activated by clicking the menu **Edit->Colors**.

Clicking the left mouse button on colored rectangle opens the standard Windows dialog box '**Color**' (Fig. 4.19). Button '**Default**' restores default colors.

A check box '**Use dotted graph grid**' enables drawing of grid with dotted line style.

We can browse the graph by moving the cursor. At the bottom of the graph the label '**Cursor:**' denotes values of the magnitude and the phase, at the cursor position.

4.9 Calibrated Measurements

In previously described measurements we have assumed that both soundcard input channels have the same sensitivity, but usually it is not true. If the difference in sensitivity of the input channels is larger than 0.2 dB, we have to make a calibration.

4.9.1 Calibration of Input Channels

The *LIMP* has a built in calibration procedure. It has to be done for a particular generator sequence length and a sampling rate.

In the calibration procedure, both input channels have to be connected to the generator output.

By clicking the menu **Record->Calibrate**, or by clicking the toolbar icon CAL, we get the '**Calibrate Input Channels**' dialog box, shown in Fig. 4.20.

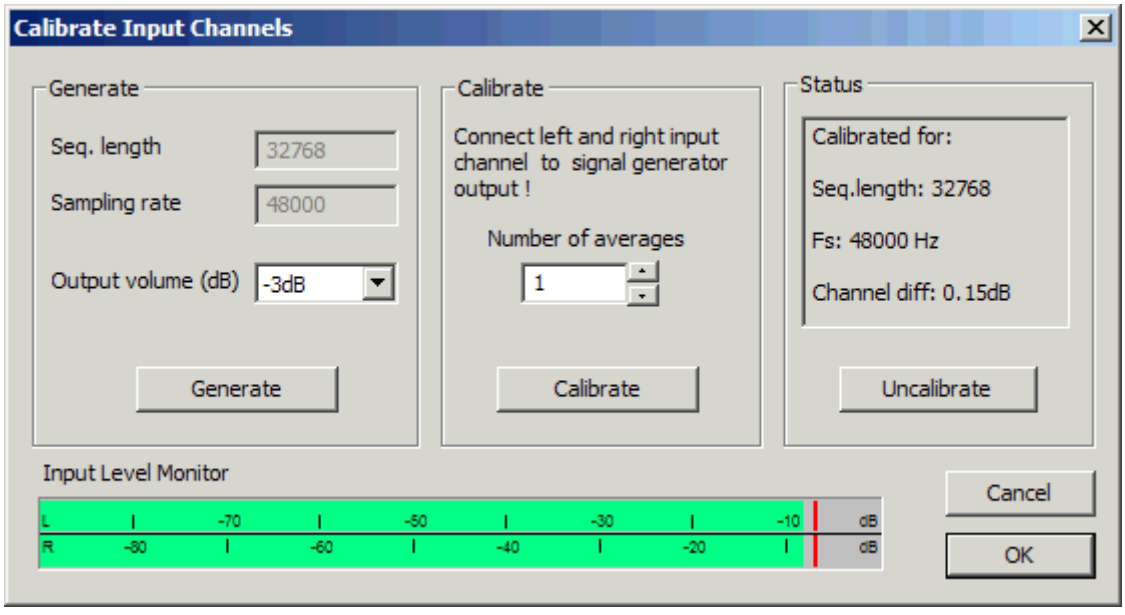


Figure 4.20 Dialog for calibration of input channel sensitivity

In section **Generate** we set the output volume for Pink PN signal of known sampling rate and sequence length. Pressing the button **'Generate'** activates the signal generation and monitoring of

input levels on the Input Level Meter. If necessary, we can adjust the output volume by using the combo box '**Output volume**'.

Finally, in section **Calibrate** we enter the **Number of averages** (recommended value is 1, as we measure high level signals) and press the button **'Calibrate'**. After a second, we have the calibrated system.

Attention: If we change the sequence length or a sampling rate, we have to repeat the calibration procedure.

The last section shows a report of the calibration procedure. The most important is information is **Channel difference** in dB. If the Channel difference is larger than 2 dB LIMP gives a warning as in Figure 4.21, and calibration is rejected.

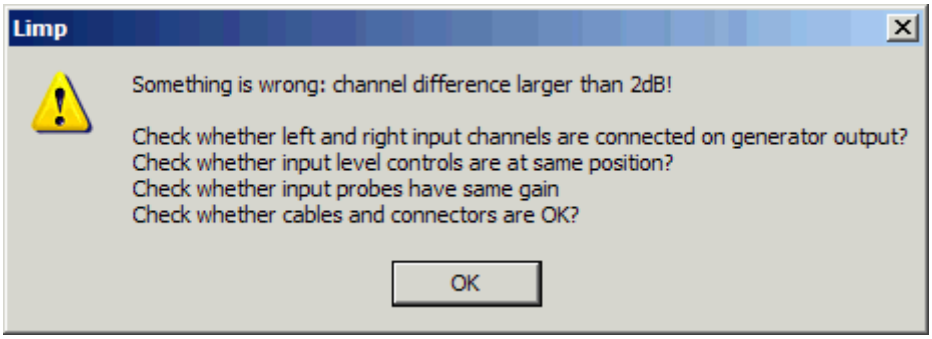

Limp
Something is wrong: channel difference larger than 2dB!
Check whether left and right input channels are connected on generator output?
Check whether input level controls are at same position?
Check whether input probes have same gain
Check whether cables and connectors are OK?
OK

Figure 4.21 Report when channel difference is larger than 2dB.

4.9.2 Cable Compensation

Sometimes we have to use long cables to connect the loudspeaker to the reference resistance. If we know the cable resistance and inductance then we can enter them in LIMP and substract from measured impedance. That procedure, called cable compensation, can be set by menu command **Setup->Cable Compensation**. It opens the dialog box shown in Fig. 4.22.

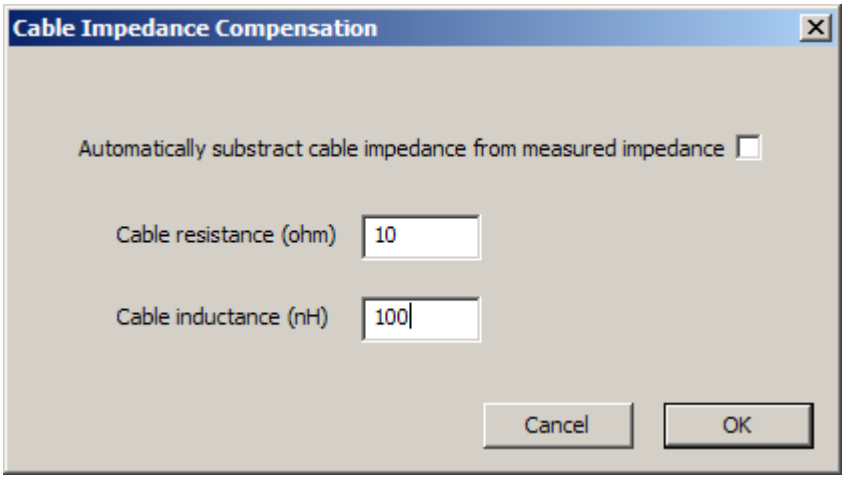


Figure 4.22 Dialog for cable impedance compensation setup

Two edit boxes are used for entering cable resistance (in ohms) and inductance (in nH). The check box "**Automatically subtract cable impedance from measured impedance**" must be checked if user wants that cable compensation be applied during the measurements.

4.8 File Manipulations

Impedance data can be saved in *LIMP* binary format (.LIM files) and in ASCII format (.TXT or .ZMA files).
Beside impedance data .LIM files can contain user defined text of arbitrary length. The user can enter any text in edit box of the **File Info** dialog box (see Fig. 4.23). This dialog box can be opened by clicking menu command '**File->Info**'.

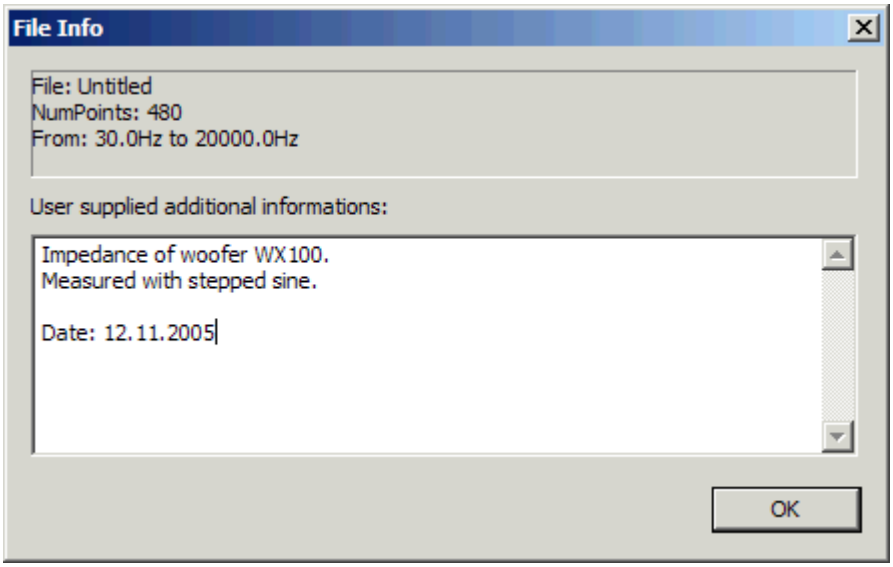


Figure 4.23 Dialog for file information

Textual files have extension .txt or .zma. They contain lines of text with three numeric literals: frequency (in Hz), magnitude (in ohms) and phase (in degrees). The line ends with new line character.

The only difference between .txt and .zma files is that .txt files can contain comment lines. Comment lines start, after arbitrary number of space or tab characters, with nonnumeric characters, while data lines start with digit or dot character.

The file can be loaded as regular impedance data or as data for the overlay plot . Only one overlay is allowed.

Binary file .LIM format is as follows:

```
char id[4];              // four characters id {'L', 'I', 'M', '\0' };
unsigned version;        // version number - started from 0x0101
unsigned reserved;       // 0
int numdata;             // number of data points
int cursorpos;           // last position of cursor
int fftlen;              // length of FFT in FFT mode
float  fs;               // sampling frequency
float data[3*numdata]    // contains  frequency, magnitude and phase
int infolength;          // info string length
char string[infolength]; // info string data
```

5 Loudspeaker Parameters

This chapter gives some definitions and measurement procedures for the estimation of loudspeaker parameters.

5.1 Definition of Physical and Dynamical Loudspeaker Parameters

An electrodynamic loudspeaker that is mounted in an infinite baffle is characterized by the following physical parameters:

Electromagnetic parameters:

R_E - voice coil DC resistance (Ω)

Z_{LE} - impedance of voice coil inductance and coupling to the pole piece (Ω) (equivalent circuits used to characterize this impedance are shown in Fig. 5.1c, and denoted as: L2R [10], [11], L3R [7], and L2RK [12], [13])

Bl - force factor (Tm)

Mechanical parameters:

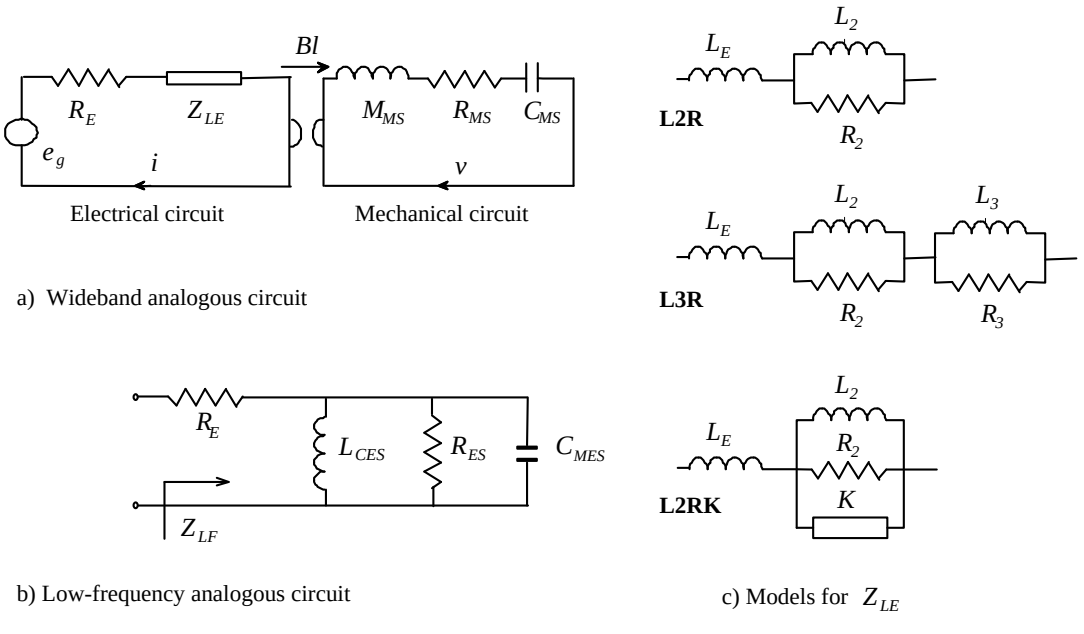
S - effective area of membrane (m^2)

C_{MS} - membrane mechanical compliance (m/N)

M_{MS} - mechanical mass of membrane plus mass of air load on membrane (kg)

R_{MS} - mechanical resistance plus membrane radiation resistance (kg/s)

(**Note:** The piston area S is normally obtained from a cone diameter measurement that includes 1/3 of the surround.)



a) Wideband analogous circuit

b) Low-frequency analogous circuit

c) Models for Z_{LE}

Figure 5.1 a) Wideband analogous circuit of an electro-dynamic loudspeaker that is mounted in an infinite baffle, b) circuit for the estimation of the low- frequency input impedance, and c) three circuit models for lossy inductor Z_{LE} (L2R, L3R and L2RK model)

Fig. 5.1 shows wideband and low-frequency equivalent circuits of an electro-dynamic loudspeaker that is mounted in a infinite baffle. The circuit for definition of low-frequency input impedance uses the following elements:

$$L_{CES} = (Bl)^2 C_{MS}, \quad R_{ES} = (Bl)^2 / R_{MS}, \quad C_{MES} = M_{MS} / (Bl)^2$$

Using these analogous circuit elements, Thiele and Small ([6],[7]) introduced *dynamical* loudspeaker parameters. They are defined in the Table 5.1.

Resonant frequency in free air (Hz)	$f_s = \frac{1}{2\pi\sqrt{M_{MS}C_{MS}}}$, $\omega_s = \frac{1}{\sqrt{M_{MS}C_{MS}}}$
Mechanical Q-factor	$Q_{MS} = \frac{\omega_S M_{MS}}{R_{MS}} = \frac{1}{\omega_S C_{MS} R_{MS}}$
Electrical Q-factor	$Q_{ES} = \frac{\omega_S M_{MS} R_E}{(Bl)^2}$
Total Q-factor	$Q_{TS} = \omega_S C_{MES} \frac{R_E R_{ES}}{R_E + R_{ES}} = \frac{Q_{MS} Q_{ES}}{Q_{MS} + Q_{ES}}$
Power available efficiency (%)	$\eta_0 = \frac{\rho_0}{2\pi c} \frac{S^2 (Bl)^2}{R_E M_{MS}^2}$
Sensitivity (1W/1m) in dB re 20uPa	$L_p(1W/1m) = 10\log\left(\frac{\rho_0 c}{2\pi}\eta_0\right) - 20\log(p_{ref}) = 112{,}1 + 10\log(\eta_0)$
Equivalent acoustical volume (m^3)	$V_{AS} = \rho_0 c^2 S^2 C_{MS}$

Table 5.1 Thiele-Small dynamical loudspeaker parameters
(ρ_0 = 1.18kg/m^3, c = 345m/s, p_{ref} = 20μPa)

Thiele and Small have shown that by using these parameters it is easy to express the low-frequency response of closed loudspeaker box as 2nd order high-pass filter and response of bass-reflex box as 4th order high-pass filter. Today, almost every loudspeaker manufacturer gives physical and Thiele-Small parameters in their loudspeaker data sheets.

An old industry standard assumed that voice coil electrical impedance contains resistive (R_E) and pure inductive ($Z_{LE} = j\omega L_E$) components. Voice coil inductance (L_E) was usually specified as inductive component of driver impedance at 1 kHz. In modern instrumentation that model is abandoned, and voice coil is treated as "lossy" inductor whose impedance Z_{LE} depends on frequency. Physically, Z_{LE} represents the input impedance of the "transformer" whose secondary winding is the pole piece. Eddy currents in the pole piece give rise to primary circuit resistance.

The simplest model [8] for a primary input impedance is $K\sqrt{j\omega} = K\,(1+j)\,\sqrt{\omega}/2$, where K is constant. It is expressed in unit called semiHenry (sH). The impedance of such semi-inductance increases with $\sqrt{\omega}$ rather thanω. This model is not practical as engineers in many numerical simulations use some form of an equivalent electrical circuit that closely matches measurement data.

The most commonly used equivalent circuit for the electrical voice coil impedance in modern instrumentation ([10], [11]) is serial connection of resistor R_E, inductor L_E and parallel connection of resistor R_2 and inductor L_2. It is denoted in Fig. 5.1 as L2R model. For many years it has been proven as useful model in many simulations.

LIMP also enables estimation of two additional and more accurate equivalent circuits for lossy inductor: L3R and L2RK models. They are also shown in Fig. 5.1. The L3R model extends L2R model with one more parallel circuit $L_3||R_3$, while in L2RK model, as defined by Thorborg [12], a semi inductance K is added parallel to circuit $L_2||R_2$.

5.2 Theory for the Estimation of Loudspeaker Parameters

For the estimation of Thiele-Small parameters (TSP), LIMP offers two methods: Thiele-Small method and Nonlinear Least Square Error (LSE) minimization method. LIMP also uses LSE minimization for estimation of lossy inductor equivalent circuit elements.

5.2.1 Estimation of Lossy Inductor Equivalent Circuit Elements

LIMP estimate values of lossy inductor equivalent circuit elements as values that give the least sum of squared differences between lossy inductor impedance $Z_{LE}(f)$ and a measured input impedance $Z_M(f)$ over all frequencies that are above the frequency of minimum of impedance. LSE error function is defined as.

$$\varepsilon = \sum_f \left[|Z_M(f)| - |Z_{LF}(f)| - |Z_{LE}(f)| \right]^2$$

where Z_{LF}(f) is a low frequency loudspeaker impedance estimated by Thiele-Small method (which is described in the next Section).
By default, LIMP estimates elements of L2R model by stable, *linearized LSE* error minimization method – sometimes also called a linear regression method. In addition, user can choose more accurate *nonlinear LSE* minimization methods for L2R, L3R and L2RK model circuit elements. For full discussion of these and TSP estimation methods it is recommended to read the paper [13].

5.2.2 Thiele-Small Method for the Estimation of TSP

It is easy to estimate Thiele-Small parameters if we have measured data for the loudspeaker impedance.

The voice coil resistance R_E should be measured with a DC ohmmeter.

At low frequencies the influence of a voice coil inductance and eddy currents is small and an expression for the loudspeaker input impedance has the following form:

$$Z(f) \cong Z_{LF} = R_E \frac{(1 - f^2 / f_S^2) + j\, f / (f_S Q_T)}{(1 - f^2 / f_S^2) + j\, f / (f_S Q_{MS})}$$

The impedance has maximum value at the resonance frequency:

$$Z_{\max} = Z(f = f_S) = R_E Q_{ES} / Q_T = R_E + R_{ES}$$

Because $Q_{ES} > Q_T$.

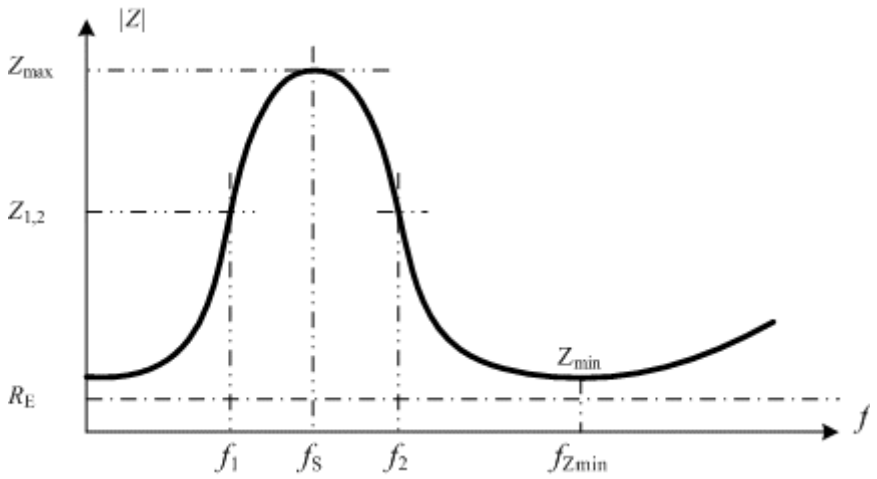


Figure 5.3 Typical impedance curve of a loudspeaker that is mounted in free air

At frequencies f_1 and f_2, (where $f_1<f_S<f_2$, $f_1 f_2 = f_S^2$) impedance values are of equal magnitude;

$$|Z(f_1)| = |Z(f_2)| = r_1 R_E, \quad \text{if} \quad f_1 f_2 = f_S^{\,2}$$

If we substitute this expression in the expression for the input impedance, we get:

$$|Z(f_{1,2})|^2 = r_1^{\,2} R_E^{\,2} = R_E^{\,2} \frac{r_0^{\,2} + (Q_{MS}(f_1 - f_2)/f_S)^2}{1 + (Q_{MS}(f_1 - f_2)/f_S)^2} .$$

From this equation we get the mechanical Q-factor:

$$Q_{MS} = \frac{f_s}{f_2 - f_1} \sqrt{\frac{r_0^{\,2} - r_1^{\,2}}{r_1^{\,2} - 1}}$$

Now we can define a step-by-step procedure for the measurement of *Q* factors:

1. Measure a voice coil resistance R_E with a dc ohm-meter
2. From impedance curve find f_S and Z_{max} . Define $r_o = Z_{max}/R_E$.
3. Choose some impedance magnitude $R_E<|Z_1|<Z_{max}$ and find both frequencies (f_1 and f_2) where $Z = Z_1$. Then define $r_1 = Z_1/R_E$.
4. Calculate Q_{MS} (from the above equation),
5. Calculate Q_{ES} using the equation $Q_{ES} = Q_{MS} / (r_o - 1)$.
6. Calculate Q_T using the equation $Q_T = Q_{ES} Q_{MS} / (Q_{ES} + Q_{MS})$.

This procedure gives good results under two conditions:

1. if we have low noise level in measurement of impedance, and
2. if we first subtract magnitude of lossy inductor impedance $|Z_{LE}|$ from the magnitude curve. In first iteration $|Z_{LE}|$ is unknown. In that case TSP estimation procedure uses values of Z_1 and f_1 from measured magnitude curve that are below the resonance frequency f_S and sets $f_2 = f_s^{\,2} / f_1$.

If we have noise in measurements we can reduce the estimation variance by repeating Thiele-Small procedure for different frequencies and average value of Q_{MS}. Good choices are frequencies where magnitude is 20% lower than maximum magnitude and 20% larger than minimum magnitude.

These two simple modifications of standard Thiele-Small procedure give accurate and reliable estimation of TSP.

5.2.3 Nonlinear LSE Minimization Method for the Estimation of TSP

Thiele Small procedure is realizable only if the value of R_E is known, but sometimes that value is not known. In that case we can use the nonlinear LSE minimization procedure where we minimize squared difference between measured impedance Z_M and model impedance Z_{LF}, i.e. we define LSE function as:

$$\varepsilon = \sum_f \left[|Z_M(f)| - |Z_{LE}(f)| - |Z_{LF}(f)| \right]^2$$

where optimization variables are R_E, f_S, Q_{MS} and Q_T. $Z_{LE}(f)$ is impedance of lossy inductor that is also estimated by LSE minimization.

If the value of R_E is known, optimization variables are: f_S, Q_{MS} and Q_T.

In some rare cases a convergent LSE minimization cannot be realized. In that case the modified standard Thiele-Small procedure would be used.

5.3 Estimation of Physical Loudspeaker Parameters

Two methods are used [7] for the estimation of physical loudspeaker parameters (M_{MS}, C_{MS} and R_{MS});

1. Added mass method,
2. Closed box method.

Both methods are implemented in the *LIMP*.

5.3.1 Added Mass Method

In this method, we first measure the impedance curve and estimate Thiele-Small parameters f_S, Q_{MS} and Q_{ES}, for the loudspeaker mounted in free air. Then we put an additional mass (M_{added}) on the membrane, measure impedance curve and estimate the shifted resonance frequency f_M and electrical Q-factor Q_{EM}. From equations for Q_{EM} and Q_{ES} we get:

$$M_{MS} = \frac{M_{added}}{\frac{f_S Q_{EM}}{f_M Q_{ES}} - 1}$$

When we know M_{MS} and f_S, it is easy to get the mechanical compliance C_{MS}, resistance R_{MS} and force factor Bl. By using equations that are defined in Table 5.1 we get

$$C_{MS} = \frac{1}{M_{MS}\omega_S^{\,2}}, \quad V_{AS} = \rho_0 c^2 S^2 C_{MS}, \quad Bl = \sqrt{\frac{\omega_S M_{MS} R_E}{Q_{ES}}}, \quad R_{MS} = \frac{\omega_S M_{MS}}{Q_{MS}}$$

5.3.2 Closed Box Method

In this method we first measure the impedance curve and estimate Thiele-Small parameters f_S, Q_{MS} and Q_{ES}, for the loudspeaker mounted in free air. Then we mount the loudspeaker in a closed box, of known volume V_B, measure impedance curve and estimate Thiele-Small parameters f_C, Q_{MC}, Q_{EC}. From these we find

$$V_{AS} = V_B\left(\frac{f_C Q_{EC}}{f_S Q_{ES}} - 1\right)$$

Then, by using equations that are defined in Table 5.1, we get

$$C_{MS} = \frac{V_{AS}}{\rho_0 c^2 S^2}, \quad M_{MS} = \frac{1}{C_{MS}\omega_S{}^2}, \quad Bl = \sqrt{\frac{\omega_S M_{MS} R_E}{Q_{ES}}}, \quad R_{MS} = \frac{\omega_S M_{MS}}{Q_{MS}}$$

5.4 Automatic Estimation of Physical and Dynamical Loudspeaker Parameters

LIMP procedures for the estimation of basic loudspeaker parameters are as follows.

To estimate Q-factors of a loudspeaker we need:

1. measured impedance data,
2. measured voice coil dc resistance (in ohms),
3. estimated membrane diameter in cm (cone diameter measurement that includes 1/3 of the surround).

By clicking the menu **Analyze->Loudspeaker parameters – Added mass method**, we get the '**Loudspeaker Parameters - Added Mass Method** ' dialog box, shown in Fig. 5.4.

In this dialog box, we enter values for **Voice coil resistance** and **Membrane diameter** in appropriate edit boxes. Press on button **'Calculate'** gets the report shown in the left edit box. This report can be pasted to the clipboard by pressing the button **'Copy'**. The click on button **Export** saves data in file using the Excel .csv format.

In the section "**Nonlinear LSE Optimization**" user sets three check boxes to allow more accurate nonlinear LSE minimization:

Estimate TSP by LSE minimization
Estimate voice coil resistance (if voice coil resistance is unavailable)
Estimate lossy inductor model

Three lossy inductor circuits:

Le+L2||R2
Le+L2||R2+ L3||R2
Le+L2||R2||K

are selectable by using the list box at the bottom of this section.
(Note: + denotes serial connection, || denotes parallel connection)

To apply NLSE method, impedance measurement have to be done in full frequency range (20kHz), or at least to frequency that is hundred times larger than loudspeaker resonance frequency.

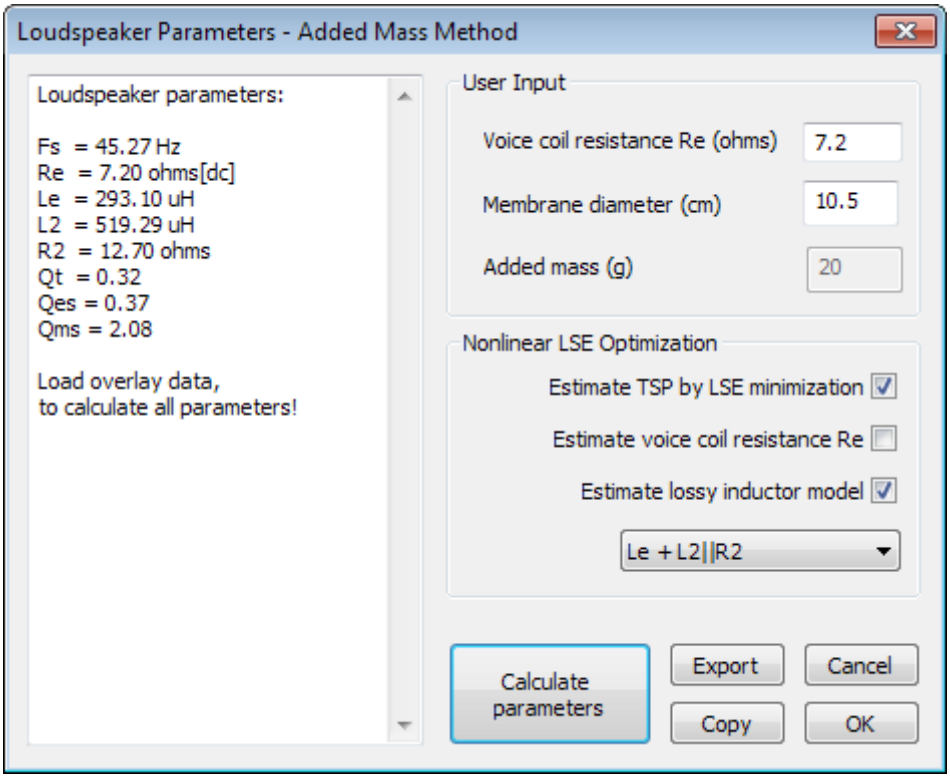


Figure 5.4 Dialog box for the estimation of loudspeaker Q-factors

5.4.1 Added Mass Method

To estimate all physical and dynamical loudspeaker parameters we must make two impedance measurements of the loudspeaker impedance in a free air. In one of the two measurements, the membrane must be loaded with an additional mass. This is what we need:

1. measured impedance data,
2. impedance data measured with an additional mass set on the membrane (as in Fig. 5.5),
3. one of the curves (1 or 2) must be set as an overlay graph,
4. measured additional mass - in grams,
5. measured voice coil dc resistance - in ohms,
6. diameter of the membrane - in cm (cone diameter that includes 1/3 of the surround),

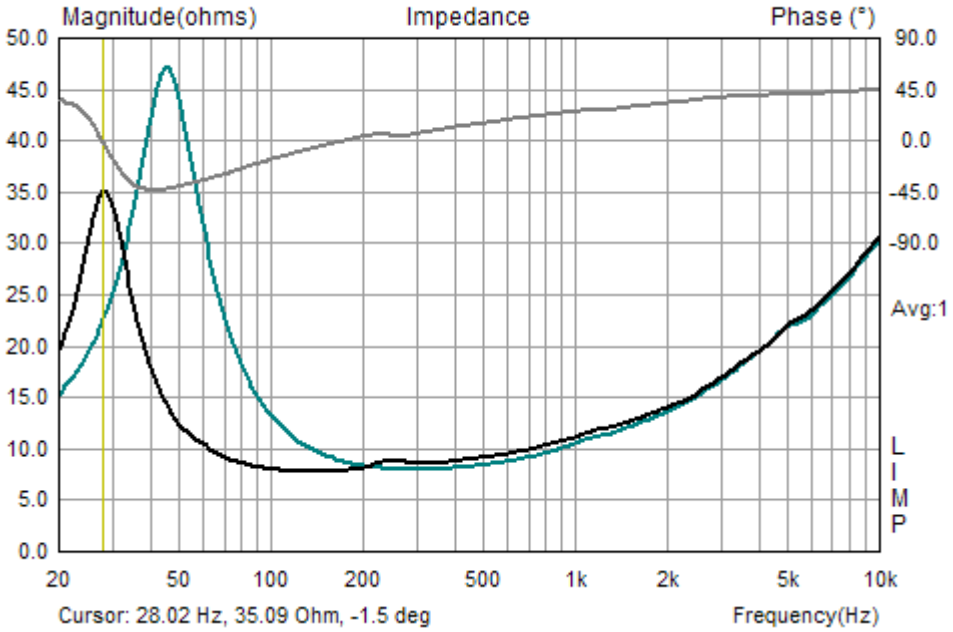


Figure 5.5 Impedance curves (the one with lower resonance frequency is obtained by adding mass to the membrane)

Then we activate the menu **Analyze->Loudspeaker parameters – Added mass method** to get the '**Loudspeaker Parameters**' dialog box, shown in Fig. 5.6.

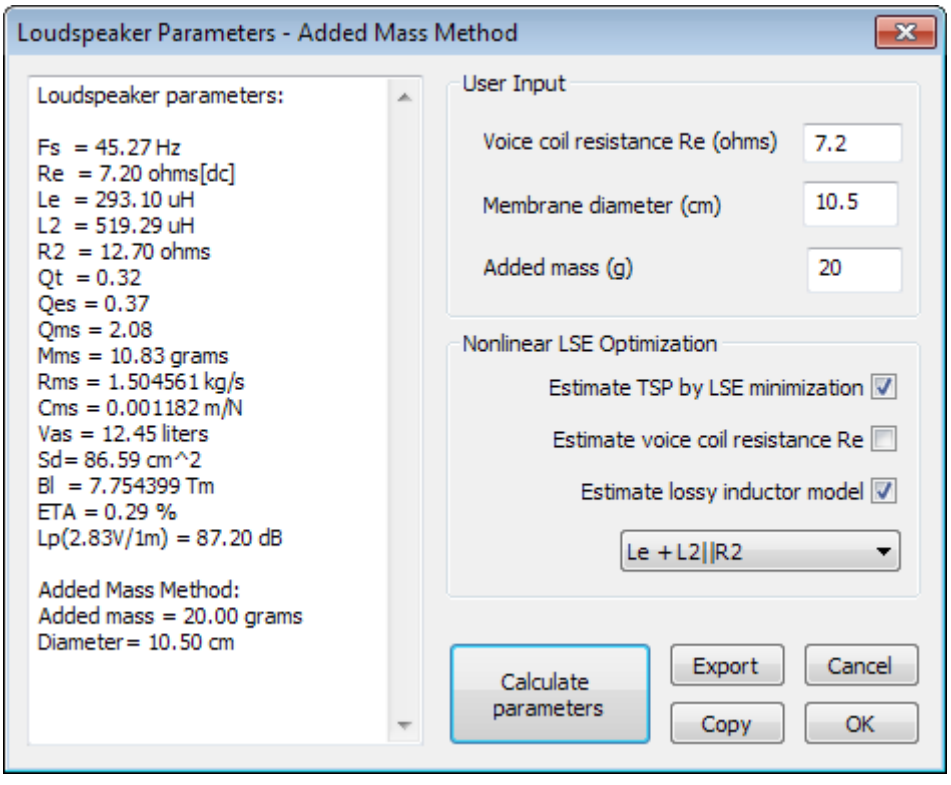


Figure 5.6 Dialog box for the estimation of loudspeaker parameters

Finally, we enter values for **Voice coil resistance, Membrane diameter** and **Added mass** in appropriate edit boxes. Press on the button **'Calculate'** gets the report shown in the left edit box. This report can be pasted to the clipboard by pressing the button **'Copy'**. The click on button **Export** saves data in file using the Excel .csv format.

In the section "**Nonlinear LSE Optimization**" user sets three check boxes to allow more accurate nonlinear LSE minimization:

Estimate TSP by LSE minimization
Estimate voice coil resistance (if voice coil resistance is unavailable)
Estimate lossy inductor model

Three lossy inductor circuits:

Le+L2||R2
Le+L2||R2+ L3||R2
Le+L2||R2||K

are selectable by using the list box at the bottom of this section.

5.4.2 Closed Box Method

To estimate all loudspeaker parameters we must have two impedance measurements. In one of measurements the loudspeaker must be mounted in a closed box of known volume. That is what we need:

1. measured impedance data,
2. impedance data measured with loudspeaker mounted in a closed box,
3. one of the curves (1 or 2) must be set as an overlay graph (as in Fig. 5.7),
4. measured value of the box volume - in liters,
5. measured voice coil dc resistance - in ohms,
6. diameter of the membrane - in cm (cone diameter that includes 1/3 of the surround),

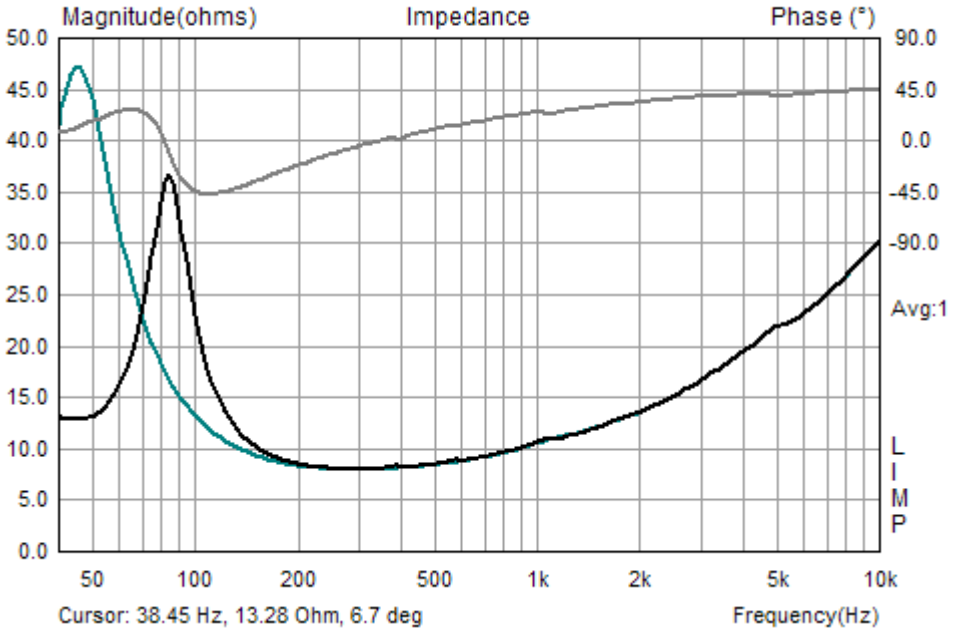


Figure 5.7 Impedance curves (the one with higher resonance frequency is obtained by mounting the loudspeaker in a closed box)

Then we activate the menu **Analyze->Loudspeaker parameters – Closed box method** to get the '**Loudspeaker parameters – Closed Box Method**' dialog box, shown in Fig. 5.8.

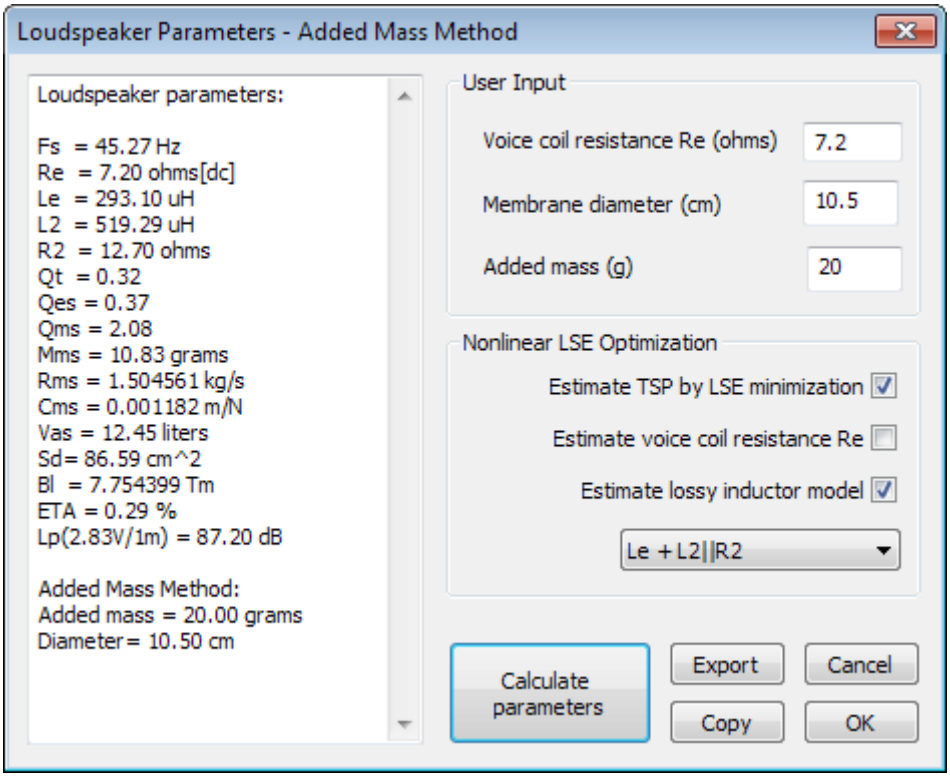


Figure 5.8 Dialog for the estimation of loudspeaker parameters using Closed Box Method

Finally, we enter values for **Voice coil resistance, Membrane diameter** and **Closed box volume** in appropriate edit boxes. Press on the button '**Calculate**' gets the report shown in the left edit box. This report can be pasted to the clipboard by pressing the button '**Copy**'. The click on button **Export** saves data in file using the Excel .csv format.

In the section "**Nonlinear LSE Optimization**" user sets three check boxes to allow more accurate nonlinear LSE minimization:

Estimate TSP by LSE minimization
Estimate voice coil resistance (if voice coil resistance is unavailable)
Estimate lossy inductor model

Three lossy inductor circuits:

Le+L2||R2
Le+L2||R2+ L3||R2
Le+L2||R2||K

are selectable by using the list box at the bottom of this section.

5.4.3 Monitoring the Quality of Loudspeaker Parameter Estimation

LIMP doesn't show any "single number" indicator of TSP estimation quality, but user can visually check the quality of estimation – by pressing F3 key in measurement window . It will generate impedance curve from currently calculated loudspeaker parameters and show it as target curve.

6 RLC Measurement

The *LIMP* can be used to measure value of resistors, capacitors and inductors, simply by calculating resistive, inductive or capacitive parts of the measured impedance. For example, Fig. 6.1 shows impedance curves of an inductor with nominal value of 110uH.

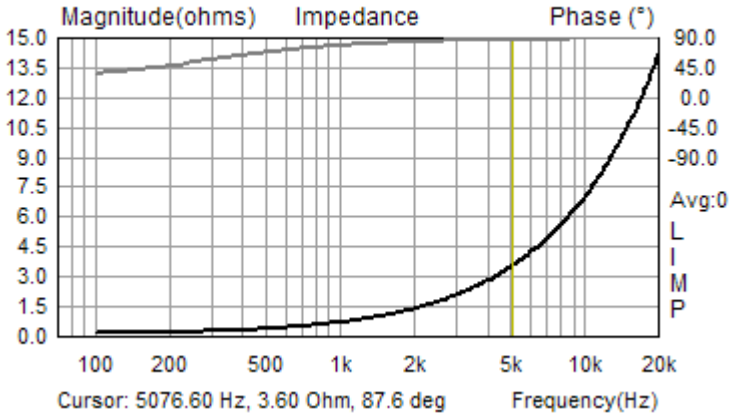


Figure 6.1 The impedance graph of a 110uH inductor

By clicking menu command **Analysis->RLC Impedance value at cursor position** we get the dialog box with report as shown on Figure 6.2. LIMP reports that measured impedance has resistive part of 0.15 ohms and imaginary part is inductive with value of 112uH.

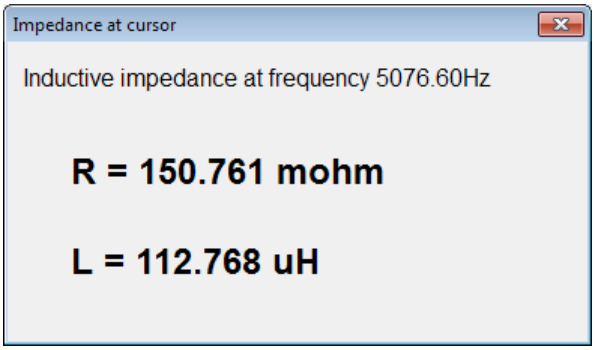


Figure 6.2 Impedance of an inductor

The LIMP measures capacitance in the same way.

6.1 Importance of Calibration

When measuring impedance and capacitance it is very important to calibrate the system before the measurement, and it is best to make the calibration with the impedance connected (DUT).

Why?

Even if there is a very small difference between channel sensitivities (i.e. 0.1dB) the LIMP can give very erroneous result, because inductor impedance has phase close to 90 degree, and capacitor impedance has phase close to minus 90 degree. In that case if there is a difference in sensitivity of measured generator voltage V1 and impedance voltage V2 (if sensitivity of probe V2 is larger than sensitivity of probe V1) estimated impedance gives phase values that are larger than 90 degree and graph shows warped jump in the phase for 180 degree. Figure 6.3 shows the case of measuring a capacitor without the calibration. In half the range the phase is close to 90 degree. It is very erroneous results as it suggest that we deal with an inductance. Figure 6.4 shows capacitor impedance after the calibration. We see correct values for phase in the whole frequency range.

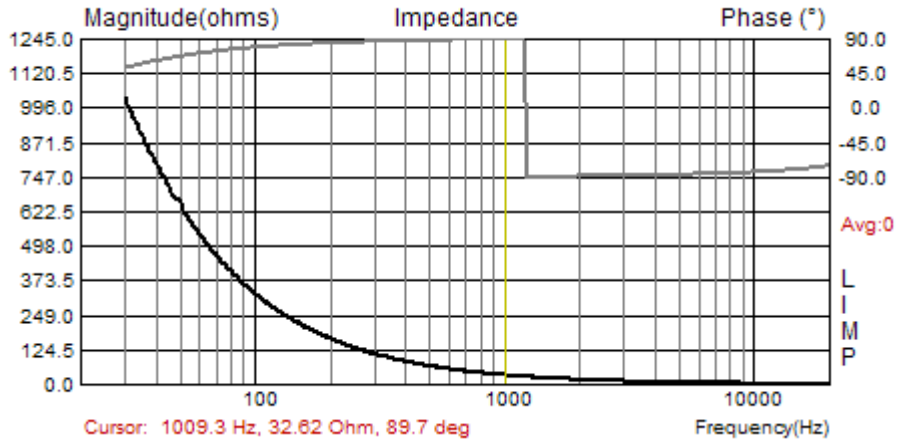


Figure 6.3 Wrongly estimated impedance of plastic capacitor 4.7uF/250V (measured without the calibration)

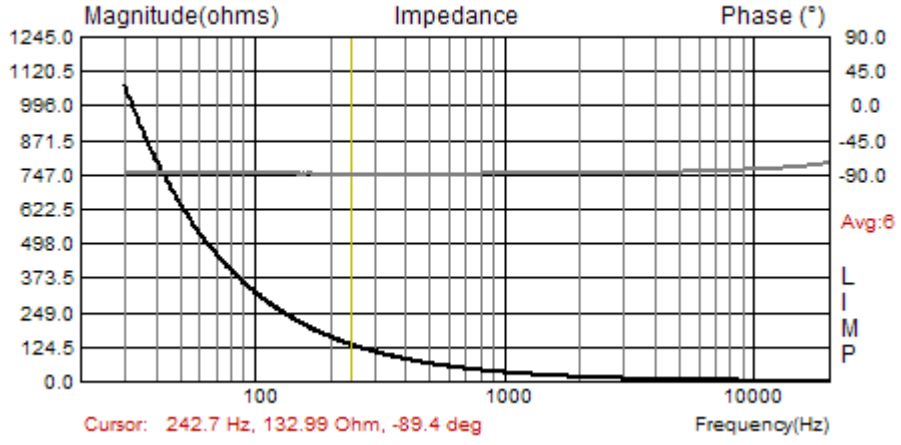


Figure 6.4 Correctly estimated impedance of plastic capacitor 4.7uF/250V (measured after the calibration)

Not all LIMP users will have this problem. As explained before, the problem exists when probe for impedance voltage V2 has higher sensitivity than the sensitivity of probe for generator voltage V1. To circumvent this condition we can change probes sensitivity or we can simply exchange input channels and also change the reference channel in the LIMP measurement setup.

Note:
To get good estimation of capacitance or inductance it is recommended to put cursor on frequency where impedance magnitude is lower than 100 ohms. It assures measurement of impedance with 1% tolerance. Why? The answer lies in fact that measured impedance is bypassed with probe or soundcard input impedance that usually has value of 10k ohms.
A lot of professional audio sound systems (like RME Fireface, EMU 0404USB, Yamaha G046) have unbalanced instrumentation inputs with input impedance in the range 470-1000kΩ. Using these systems, a much better accuracy is possible.

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